A1

The present invention provides a two-fold approach to sound quality improvement under high noise situations and its practical implementation in a hearing aid. In one aspect, the present invention removes noise from the input signal and controls a compression stage with a cleaner signal, compared to the use of the original noisy input signal for controlling compression as is done in the prior art. The signal for amplification is, optionally, processed with a different noise reduction algorithm. Under certain circumstances, it may be desirable to use the same noise reduced signal for application and compression control in which case the two noise reduction blocks merge. In another instance, it may be desirable to alter or eliminate the noise reduction in the upper path.

Please amend the paragraph beginning on page 2, line 18 as follows:

A2

Sub B1

In accordance with a first aspect of the present invention, there is provided a method of reducing noise in a signal containing speech and having a signal to noise ratio, the method comprising the steps:

(1) detecting the presence and absence of speech;

(2) in the absence of speech, determining a noise magnitude spectral estimate $(|\hat{N}(f)|)$;

(3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $(|\hat{N}(f)|)$;

(4) calculating an attenuation function ($H(f)$) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $(|\hat{N}(f)|)$, the attenuation function ($H(f)$) being dependent on the signal to noise ratio; and,

(5) modifying the input signal by the attenuation function ($H(f)$), to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

Please insert the following paragraph before the paragraph beginning on page 2, line 28 as follows:

A3

Sub B2

Preferably, the method further comprises the steps of:

(6) supplying the input signal to an amplification unit;

(7) providing the noise reduced signal to a compression circuit which generates a control input for the amplification unit; and,

(8) controlling the amplification unit with the control signal to modify the input signal to generate an output signal with compression and reduced noise.

Advantageously, step (7) comprises subjecting the input signal to an auxiliary noise reduction algorithm to generate an auxiliary noise reduced signal and providing the auxiliary noise reduced signal to the compression circuit.

Please delete the paragraph beginning on page 2, line 28.

Please insert the following paragraphs before the paragraph beginning on page 2, line 32 as follows:

A4

Sub B3

In one embodiment, step (6) comprises applying the steps (1) to (5) to the input signal prior to supplying the input signal to the amplification unit.

Furthermore, in one embodiment, the input signal may be subjected to a main noise reduction algorithm, to generate a modified input signal, which is supplied to the amplification unit. the auxiliary noise reduction algorithm may comprise the same noise reduction method as the main noise reduction algorithm. Alternatively, the auxiliary noise reduction algorithm may be different from the noise reduction method in the main noise reduction algorithm.

Please delete the paragraph beginning on page 2, line 32.

Please amend the paragraph beginning on page 3, line 12 as follows:

A5

Conveniently, the square of the speech magnitude spectral estimate $\left(\left|\hat{S}(f)\right|\right)$ may be determined by subtracting the square of the noise magnitude spectral estimate

$\left(\left|\hat{N}(f)\right|\right)$ from the square of the magnitude spectrum of the input signal (|X(f)|). In a preferred embodiment, the attenuation factor is a function of frequency and is calculated in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta |\hat{N}(f)|^2}{|X(f)|^2}\right]^\alpha$$

A5

where f denotes frequency, H(f) is the attenuation function, |X(f)| is the magnitude spectrum of the input audio signal; $\left(\left|\hat{N}(f)\right|\right)$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function. The oversubtraction factor β is, preferably, varied as a function of the signal to noise ratio, with β being zero for high and low signal to noise ratios and with β being increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios β decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

Please amend the paragraph beginning on page 3, line 25 as follows:

A4

Advantageously, the oversubtraction factor β is divided by a preemphasis function of frequency P(f) to give a modified oversubtraction factor $\hat{\beta}(f)$, the preemphasis function being such as to reduce $\hat{\beta}(f)$ at high frequencies, to reduce attenuation at high frequencies.

Please amend the paragraph beginning on page 4, line 12 as follows:

A1

Another aspect of the present invention provides for a method of detecting the presence or the absence of speech in an audio signal, the method comprising taking a block of the audio signal and performing an auto-correlation on that block to form a correlated signal; and checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for speech.

Please insert the following paragraphs after the paragraph beginning on page 4, line 18 as follows:

In a further aspect the present invention provides an apparatus for reducing noise in an input signal, the apparatus including an input for receiving the input signal. The apparatus comprises a compression circuit for receiving a compression control signal and generating an amplification control signal in response, and an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise. The apparatus further comprises an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal.

The apparatus may further comprise a main noise reduction unit connected to the input for generating a noise reduced signal and supplying the noise reduced signal in place of the input signal to the amplification unit.

A8

Preferably, the input signal contains speech and the main noise reduction unit comprises a detector connected to the input and providing a detection signal indicative of the presence of speech and a magnitude means for determining the magnitude spectrum of the input signal (|X(f)|), with both the detector and the magnitude means being connected to the input of the apparatus. The main noise reduction unit further comprises a spectral estimate means for generating a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and being connected to the detector and to the input of the apparatus, a noise filter calculation unit connected to the spectral estimate means and the magnitude means, for receiving the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and magnitude spectrum of the input signal (|X(f)|) and calculating an attenuation function ($H(f)$), and a multiplication unit coupled to the noise filter calculation unit and the input signal for producing the noise reduced-signal.

Please delete the paragraph beginning on page 4, line 18.

Please amend the paragraph beginning on page 5, line 1 as follows:

A9

Figure 1 is a conceptual blocked diagram for hearing aid noise reduction and compression;

Please amend the paragraph beginning on page 5, line 16 as follows:

A10

Here, the position of the noise reduction unit 18 can advantageously provide a cleaner signal for controlling the compression stage. The noise reduction unit 18 provides a first generating means which generates an auxiliary signal from an auxiliary noise reduction algorithm. The auxiliary algorithm performed by unit 18 may be identical to the one performed by unit 16, except with different parameters. Since the auxiliary noise reduced signal is not heard, unit 18 can reduce noise with increased aggression. This auxiliary signal, in turn, controls the compression circuitry 20, which comprises second generating means for generating a control input for controlling the amplification unit 22.

Please amend the paragraph beginning on page 6, line 1 as follows:

A11

With reference to Figure 2, this shows a block diagram of a specific realization of the proposed noise reduction technique which is preferably carried out by noise reduction unit 18 (and possibly also noise reduction unit 16). The incoming signal at 10 is first blocked and windowed, as detailed in applicant's simultaneously filed international application serial no. PCT/CA98/00329 corresponding to international publication no. WO 98/47313 which is incorporated herein by reference. The blocked and windowed output provides the input to the frequency transform (all of these steps take place, as indicated, at 32), which preferably here is a Discrete Fourier Transform (DFT), to provide a signal X(f). The present invention is not however restricted to a DFT and other transforms can be used. A known, fast way of implementing a DFT with mild restrictions on the transform size is the Fast Fourier Transform (FFT). The input 10 is also connected to a speech detector 34 which works in parallel to isolate the pauses in the incoming speech. For simplicity, reference is made here to "speech", but it will be understood that this encompasses any desired audio signal, capable of being isolated or detected by detector 34. These pauses provide opportunities to update the noise